Amendments to the Claims:

This listing of claims will replace all prior versions, and listings, of claims in the application:

Listing of Claims:

1. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting [[to]] between the public switched telephone network and an IP network medium; and

~~an IP network medium connected to the gateway;~~

a network server in communication with the IP network medium for automated interaction with a user participating in the call; ~~and~~

~~a configuration server for performing a blasting process to provide automated dynamic management of the network server~~ where the call is transmitted as a data stream of uncompressed data from the gateway to the network server.

2. (original) The voice response system of claim 1, wherein the network server comprises a host computer for executing a voice application program, a grammar database corresponding to a set of recognizable utterances, and a voice recognition engine for comparing a speech input from the user against the set of recognizable utterances.

3. (original) The voice response system of claim 2, wherein the voice application program is a VoiceXML program.

4. (currently amended) The voice response system of claim 2, further comprising a firewall in communication with the IP network medium for connecting the network server to an external IP network through the firewall, wherein the voice application program is remotely hosted on the external IP network.

5. (original) The voice response system of claim 2, wherein the network server performs call control communications with the PSTN-to-IP gateway in accordance with a SIP protocol.

6. (currently amended) A ~~scalable, computerized, Internet protocol (IP) based~~ voice response system for servicing a plurality of calls received over a public switched telephone network (PSTN), comprising:

a PSTN-to-IP gateway for connecting to the ~~public switched telephone network~~ PSTN and an IP network medium;

~~an IP network medium connected to the gateway;~~

a plurality of network servers, in communication with the IP network medium and located in close physical proximity to the PSTN-to-IP gateway, for automated interaction with a set of users participating in the plurality of calls; and

a proxy server in communication with the PSTN-to-IP gateway for load balancing the plurality of calls and providing differentiated and targeted service control for the plurality of calls amongst the plurality of network servers.

7. (original) The voice response system of claim 6, wherein each network server of the plurality of network servers comprises a host computer having a distinct network identification number.

8. (original) The voice response system of claim 7, further comprising a configuration server for automatically loading and configuring an initial software environment for the host computer during its initial bootup sequence based upon the network identification number.

9-15. (canceled)

16. (currently amended) The voice response system of claim 1, further including a proxy server~~, wherein the configuration server directs the proxy server how~~ configured to perform call discrimination.

17. (previously presented) The voice response system of claim 16, wherein call discrimination can allow calls on an entity basis.

18. (previously presented) The voice response system of claim 16, wherein call discrimination can disallow calls on an entity basis.

19. (currently amended) The voice response system of claim 16, ~~wherein the~~ further comprising a configuration server ~~provides~~ to perform re-purposing of at least one of the proxy server [[and]] or the network server.

20. (currently amended) The voice response system of claim 1, further including a proxy server, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the proxy server follows at least one predetermined call routing rule ~~provided by the configuration server~~ to process the call.

21. (currently amended) The voice response system of claim 20, wherein the at least one predetermined call routing ~~rules include~~ rule includes sending a busy signal to a first entity and allowing calls [[to]] from a second entity.

22. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing ~~a call~~ calls received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network and an IP network medium;

~~an IP network medium connected to the gateway;~~

a network server in communication with the IP network medium for automated interaction with ~~a user~~ users participating in the ~~call~~ calls; and

a proxy server in communication with the IP network medium and the network server~~,~~ ~~wherein the proxy server provides differentiated and targeted service control for the call~~ to allow or disallow the calls based on a telephone number associated with the calls.

23. (canceled)

24. (canceled)

25. (currently amended) The voice response system of claim 22, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the ~~differentiated and targeted service control includes following~~ proxy server is configured to process the calls based on predetermined call routing rules.

26. (currently amended) The voice response system of claim 25, wherein the predetermined call routing rules include sending a busy signal to a first ~~client~~ one of the users and allowing calls [[to]] from a second ~~client~~ one of the users.

27. (previously presented) The voice response system of claim 22, wherein the network server and proxy server are dynamically reconfigurable.

28. (previously presented) The voice response system of claim 27, wherein dynamic reconfiguration includes mapping a new software configuration.

29. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network and an IP network medium;

~~an IP network medium connected to the gateway;~~

a network server, in communication with the IP network medium and located in close physical proximity to the gateway, for automated interaction with a user participating in the call; and

means for providing differentiated and targeted service control over the call in operative relation to the IP network medium and the network server.

30. (previously presented) The voice response system of claim 29, wherein the differentiated and targeted service control can allow calls on a per client basis.

31. (previously presented) The voice response system of claim 29, wherein the differentiated and targeted service control can disallow calls on a per client basis.

32. (previously presented) The voice response system of claim 29, wherein if a number of calls exceeds a predetermined threshold, then the means for providing the differentiated and targeted service control follows predetermined call routing rules.

33. (currently amended) The voice response system of claim 32, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls [[to]] from a second client.

34. (previously presented) The voice response system of claim 29, wherein the means for providing the differentiated and targeted service control can be dynamically reconfigured.

35. (currently amended) The voice response system of claim 34, wherein dynamic reconfiguration includes a new setup of [[the]] a proxy server.

36. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network and an IP network medium;

~~an IP network medium connected to the gateway;~~

a network server in communication with the IP network medium for automated interaction with a user participating in the call;

a configuration server for providing automated dynamic management of the network server; and

a proxy server, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the proxy server follows at least one predetermined call routing rule ~~provided by the configuration server~~ to process the call.

37. (currently amended) The voice response system of claim 36, wherein the at least one predetermined call routing ~~rules include~~ rule includes sending a busy signal to a first entity and allowing calls [[to]] from a second entity.

38. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network and an IP network medium;

~~an IP network medium connected to the gateway;~~

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

a proxy server in communication with the IP network medium and the network server, wherein the proxy server provides call discrimination,

wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the call discrimination includes following predetermined call routing rules.

39. (currently amended) The voice response system of claim 38, wherein the predetermined call routing rules include ~~sending a busy signal to~~ disallowing a call from a first client and allowing ~~calls to~~ a call from a second client.

40. (currently amended) A ~~computerized, Internet protocol (IP) based~~ voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network and an IP network medium;

~~an IP network medium connected to the gateway;~~

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

means for providing call discrimination over the call in operative relation to the IP network medium and the network server,

wherein if a number of calls exceeds a predetermined threshold, then the means for providing call discrimination follows predetermined call routing rules.

41. (currently amended) The voice response system of claim 40, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls [[to]] from a second client.

42. (withdrawn) A call handling system comprising:

a packet network;

a network interface for interfacing the packet network to at least one external network; and

an application server for sending signaling information to the network interface via the packet network, the signaling information indicating a call handling function for a call received by the network interface.

43. (withdrawn) The call handling system of claim 42, wherein the network interface further comprises means for transmitting the signaling information to the at least one of the external networks to execute the call handling function.

44. (withdrawn) The call handling system of claim 42, wherein the call handling function includes one of an outbound routing of the call, an outbound transfer of the call and a rejection of the call.

45. (withdrawn) The call handling system of claim 42, wherein the network interface further comprises means for selecting the application server from a plurality of application servers.

46. (withdrawn) The call handling system of claim 45, wherein the means for selecting the application server comprises a SIP proxy server.

47. (withdrawn) The call handling system of claim 45, wherein the means for selecting the application server comprises means for establishing a session with the application server and establishing a media stream between the application server and the network interface.

48. (withdrawn) The call handling system of claim 47, wherein the media stream is controlled by the real time transport protocol ("RTP").

49. (withdrawn) The call handling system of claim 42, wherein the call handling function is an outbound transfer of the call and the network interface establishes a connection with a third party as a function of the signaling information.

50. (withdrawn) The call handling system of claim 42, wherein the packet network is a voice over Internet protocol ("VoIP") network.

51. (withdrawn) The call handling system of claim 42, wherein the signaling information is generated using the session initiation protocol ("SIP").

52. (withdrawn) The call handling system of claim 42, wherein the at least one. external network is a TDM ("Time Division Multiplexing") network.

53. (withdrawn) The call handling system of claim 42, wherein the at least one external network is a second packet network.

54. (withdrawn) A method operating a call handling system, the method comprising:

receiving a call at a network interface from an external network;

selecting an application server to handle the call; and

sending a first set of signaling information from the application server to the network interface via a packet network, the first set of signaling information indicating a call handling function for the call.

55. (withdrawn) The method of claim 54, further comprising transmitting a second set of signaling information from the network interface to the external network in response to the first set of signaling information, the second set of signaling information causing the call handling function to be executed on the call.

56. (withdrawn) The method of claim 54, wherein selecting the application server comprises sending a second set of signaling information from the network interface to the application server via the packet network.

57. (withdrawn) The method of claim 54, wherein the call handling function includes one of an outbound routing of the call, an outbound transfer of the call and a rejection of the call.

58. (withdrawn) The method of claim 54, further comprising selecting the application server comprises selecting the application server from a plurality of application servers using a SIP proxy server.

59. (withdrawn) The method of claim 54, wherein selecting the application server comprises:

establishing a session with the application server; and

establishing a media stream between the application server and the network interface.

60. (withdrawn) The method of claim 59, wherein the media stream is controlled by the real time transport protocol ("RTP").

61. (withdrawn) The method of claim 54, wherein the call handling function is an outbound transfer of the call, and wherein the method further comprises establishing a connection between the network interface and a third party as a function of the first set of signaling information.

62. (withdrawn) The method of claim 54, wherein the packet network is a voice over Internet protocol ("VoIP") network.

63. (withdrawn) The method of claim 54, wherein sending the first set of signaling information comprises generating the first set of signaling information using the session initiation protocol ("SIP").

64. (withdrawn) The method of claim 54, wherein the external network is a TDM ("Time Division Multiplexing") network.

65. (withdrawn) The method of claim 54, wherein the external network is a packet network.

66. (withdrawn) A call handling system comprising:

a packet network;

a network interface for receiving a call from an external network and generating a first set of signaling information in response to the call;

an application server for receiving the first set of signaling information via the packet network, for establishing a session with the network interface in response to the first set of signaling information, and for sending a second set of signaling information to the network interface via the packet network, the second set of signaling information indicating a call handling function to be performed on the call.

67. (new) A system for processing a call received over a public switched telephone network (PSTN), comprising:

a PSTN-to-IP gateway connected to receive the call from the PSTN;

a network server in communication with the gateway for interacting with a user participating in the call; and

an IP communication path of approximately 100 meters or less in length for connecting the gateway to the network server.

68. (new) The voice response system of claim 29, wherein the call is transmitted as uncompressed data on the IP network medium between the gateway and the network server.